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EXAMINER	
CHAWAN, VIJAY B	
ART UNIT	**PAPER NUMBER**
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Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 10/03)

Office Action Summary	Application No. 09/866,585	Applicant(s) RAJAN, JEBU JACOB	
	Examiner Vijay B. Chawan	Art Unit 2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 4/15/05.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-68 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-25,37-61,64,67 and 68 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☒ Claim(s) 26-36,62,63 and 65 are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1)☒ Notice of References Cited (PTO-892)

2)☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3)☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date see attached.

4)☐ Interview Summary (PTO-413) Paper No(s)/Mail Date._____.

5)☐ Notice of Informal Patent Application (PTO-152)

6)☐ Other: _____.

DETAILED ACTION

Election/Restrictions

1. Applicant's election with traverse of Group I, claims 1-25, 37-61, 64, 67, and 68 in the reply filed on 4/15/2005 is acknowledged. The traversal is on the ground(s) that the two groups of claims are closely related and that a proper search of any one group would likely include a search of the claims of the other group. This is not found persuasive because, Group I invention does have separate utility such as audio encoding using probability density functions. See MPEP § 806.05(d).

The requirement is still deemed proper and is therefore made FINAL.

Information Disclosure Statement

2. The second Information Disclosure Statement/list (specifically a PTO-1449 is missing. The IDSs' that were considered are 1, 3-9. The IDSs were titled First Information Disclosure Statement and so on until the Ninth Information Disclosure Statement of which, the Second Information Disclosure Statement is missing.

Double Patenting

3. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. See *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970); and, *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent is shown to be commonly owned with this application. See 37 CFR 1.130(b).

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

4. Claims 1-25, 37-61, 64, 67, and 68 are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-66 of copending Application No. 09/866,854. Although the conflicting claims are not identical, they are not patentably distinct from each other because the claims of the instant application are substantially similar to claims 1-66 of Application serial number 09/866,584.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

6. Claims 1-25, 37-61, 64, 67, 68 are rejected under 35 U.S.C. 102(b) as being anticipated by Scholz et al., (5,325,397).

As per claim 1, Scholz et al., teach an audio encoding system comprising:

a memory for storing a predetermined function which gives, for a given set of audio signal values, a probability density for parameters of a predetermined audio model which is assumed to have generated the set of audio signal values, the probability density defining, for a given set of model parameter values, the probability that the predetermined audio model has those parameter values, given that the model is assumed to have generated the set of audio signal values (Col.11, line 67 – Col.12, line 7);

means for receiving a set of audio signal values representative of an input audio signal (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22);

means for applying the set of received audio signal values to said stored function to give the probability density for said model parameters for the set of received audio signal values (Col.10, lines 1-18, fig.2);

means for processing said function with said set of received audio signal values applied, to derive samples of parameter values from said probability density (Col.12, lines 34-41, Col.10, lines 31-39);

means for analyzing at least some of said derived samples of parameter values to determine parameter values that are representative of the set of received audio signal values (Col.12, lines 42-48, Col.11, lines 13-49); and,

means for encoding said determined parameter values to generate encoded data representative of the received audio signal values (Col.12, lines 42-48, Col.11, lines 13-49).

As per claim 2, Scholz et al., teach a system according to claim 1, wherein said processing means is operable to draw samples iteratively from said probability density function (Col.7, lines 1-9).

As per claim 3, Scholz et al., teach a system according to claim 2, wherein said processing means comprises a Gibbs sampler (Gibbs sampling is a known scheme for estimating a time-varying autoregressive process for audio signals with a flexible model).

As per claim 4, Scholz et al., teach a system according to claim 2, wherein said analyzing means is operable to determine a histogram of said drawn samples and wherein said values of said parameters are determined from said histogram (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 5, Scholz et al., teach a system according to claim 4, wherein said processing means is operable to determine said values of said first parameters using a

weighted sum of said drawn samples and wherein the weighting for each sample is determined from said histogram (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 6, Scholz et al., teach a system according to claim 1, wherein said receiving means is operable to receive a sequence of sets of signal values representative of an input audio signal, and wherein said applying means, processing means and analyzing means are operable to perform their function with respect to each set of received audio signal values to determine parameter values that are representative of each set of received audio signal values (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 7, Scholz et al., teach a system according to claim 6, wherein said processing means is operable to use the values of parameters obtained during the processing of a preceding set of signal values as initial estimates for the values of the corresponding parameters for a current set of signal values being processed (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 8, Scholz et al., teach a system according to claim 6, wherein said sets of signal values in said sequence are non-overlapping (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 9, Scholz et al., teach a system according to claim 6, wherein said processing means comprises means for varying the number of parameters used to represent the audio signal within each set of audio signal values (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 10, Scholz et al., teach a system according to claim 1, wherein said audio model comprises an auto-regressive process model and wherein said parameters include auto-regressive model coefficients (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 11, Scholz et al., teach a system according to claim 1, wherein said received set of audio signal values are representative of an input speech signal (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 12, Scholz et al., teach a system according to claim 11, wherein said received set of speech signal values are representative of a speech signal generated by a speech source as distorted by a transmission channel between the speech source and the receiving means, wherein said predetermined function includes a first part having first parameters which models said source and a second part having second parameters which models said channel, wherein said processing means is operable to derive samples of at least said first parameters, and wherein said analyzing means is operable to determine values of said first parameters that are representative of said speech generated by said speech source before it was distorted by said transmission channel (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 13, Scholz et al., teach a system according to claim 12, wherein said function is in terms of a set of raw speech signal values representative of speech generated by said source before being distorted by said transmission channel, wherein the system further comprises second processing means for processing the received set of signal values with initial estimates of said first and second parameters, to generate an estimate of the raw speech signal values and wherein said applying means is operable

to apply said estimated set of raw speech signal values to said function in addition to said set of received signal values (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 14, Scholz et al., teach a system according to claim 13, wherein said second processing means comprises a simulation smoother (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 15, Scholz et al., teach a system according to claim 13, wherein said second processing means comprises a Kalman filter (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 16, Scholz et al., teach a system according to claim 12, wherein said second part is a moving average model and said second parameters comprise moving average coefficients (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 17, Scholz et al., teach a system according to claim 1, further comprising means for evaluating said probability density function for the set of received signal values using one or more of said drawn samples of parameter values for different numbers of parameter values, to determine respective probabilities that the predetermined signal model has those parameter values and wherein said processing means is operable to process at least some of said drawn samples of parameter values and said evaluated probabilities to determine said values of said parameters that are representative of the received audio signal (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 18, Scholz et al., teach a system according to claim 1, wherein said analyzing means is operable to analyze at least some of said derived samples of parameter values to determine a measure of the variance of at least some of said samples of parameter values, wherein said system further comprises means for determining an indication of the quality of the received audio signal using said variance measure, and wherein said encoding means is operable to encode said determined parameter values in dependence upon the determined quality indication (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 19, Scholz et al., teach a system according to claim 18, wherein said encoding means is operable to encode said parameter values using a first encoding technique if said quality indication is above a predetermined value and is operable to encode said parameter values using a second encoding technique if said quality indication is below said value (Col.18, line 63 – Col.19, line 2).

As per claim 20, Scholz et al., teach a system according to claim 19, wherein said first encoding technique is operable to minimize the data to be transmitted and wherein said second encoding technique is operable to minimize information lost in the encoding (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 21, Scholz et al., teach an audio transmission system comprising, a transmission unit comprising means for receiving an audio signal, and audio signal encoding system according to claim 1, for generating encoded parameter values representative of received audio signal values, and means for transmitting the encoded parameter values, and a receiver unit comprising means for receiving the transmitted

parameter values, and means for processing the received parameter values to generate an output signal in dependence thereon (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 22, Scholz et al., teach a system according to claim 21, wherein said processing means of said receiving unit comprises speech synthesis means for generating synthesized speech signal in dependence upon the received parameter values (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 23, Scholz et al., teach a system according to claim 21, wherein said processing means of said receiving unit comprises a speech recognition system whish is operable to compare the received parameter values with stored reference models to generate a recognition result (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 24, Scholz et al., teach a system according to claim 21, wherein said transmission unit is operable to transmit said quality indication to said receiving unit and wherein said receiving unit is operable to receive said quality indication and to decode said encoded parameters in dependence upon the received quality indication (Col.18, line 62 – Col.19, line 2).

As per claim 25, Scholz et al., teach a system according to claim 24, wherein said receiving unit is operable to decode said encoded parameter values in accordance with a first decoding technique if said quality indication has a value above a predetermined threshold value and is operable to decode said encoded parameter

values in accordance with a second decoding technique if said quality indication is below said predetermined value (Col.18, line 62 – Col.19, line 2).

As per claim 37, Scholz et al., teach an audio encoding method comprising the steps of:

storing predetermined function which gives, for a given set of audio signal values, a probability density for parameters of a predetermined audio model which is assumed to have generated the set of audio signal values, the probability density defining, for a given set of model parameter values, the probability that the predetermined audio model has those parameter values, given that the model is assumed to have generated the set of audio signal values (Col.11, line 67 – Col.12, line 7);

receiving a set of audio signal values representative of an input audio signal at a receiver (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22);

applying the set of received audio signal values to said stored function to give the probability density for said stored function to give the probability density for said model parameters for the set of received audio signal values (Col.10, lines 1-18, fig.2);

processing said function with said set of received audio signal values applied, to derive samples of parameter values from said probability density (Col.12, lines 34-41, Col.10, lines 31-39);

analyzing at least some of said derived samples of parameter values to determine parameter values that are representative of the set of received audio signal values (Col.12, lines 42-48, Col.11, lines 13-49); and,

encoding said determined parameter values to generate encoded data representative of the received audio signal values (Col.12, lines 42-48, Col.11, lines 13-49).

As per claim 38, Scholz et al., teach the method according to claim 37, wherein said processing step draws samples iteratively from said probability density function (Col.7, lines 1-9).

As per claim 39, Scholz et al., teach the method according to claim 38, wherein said processing step uses the Gibbs sampler (Gibbs sampling is a known scheme for estimating a time-varying autoregressive process for audio signals with a flexible model).

As per claim 40, Scholz et al., teach the method according to claim 38, wherein said analyzing step determines a histogram of said drawn samples and wherein said values of said parameters are determined from said histogram (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 41, Scholz et al., teach the method according to claim 40, wherein said processing step determines said values of said first parameters using a weighted sum of said drawn samples and wherein the weighting for each sample is determined from said histogram (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 42, Scholz et al., teach the method according to claim 37, wherein said receiving step receives a sequence of sets of signal values representative of an input audio signal and wherein said applying step, processing step and analyzing step are performed for each set of received audio signal values to determine parameter

values that are representative of each set of received audio signal values (Col.11, lines 13-49, Col.10, lines 31-40).

As per claim 43, Scholz et al., teach the method according to claim 42, wherein said processing step uses the values of parameters obtained during the processing of a preceding set of signal values as initial estimates for the values of the corresponding parameters for a current set of signal values being processed (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 44, Scholz et al., teach the method according to claim 42, wherein said sets of signal values in said sequence are non-overlapping (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 45, Scholz et al., teach the method according to claim 42, wherein said processing step comprises the step of varying the number of parameters used to represent the audio signal within each set of audio signal values (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 46, Scholz et al., teach the method according to claim 37, wherein said audio model comprises as auto-regressive process model and wherein said parameters include auto-regressive model coefficients (Col.10, lines 31-40, Col.12, lines 41-48).

As per claim 47, Scholz et al., teach the method according to claim 37, wherein said received set of said audio signal values are representative of an input speech signal (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 48, Scholz et al., teach the method according to claim 47, wherein said received set of speech signal values are representative of a speech signal generated by a speech source as distorted by a transmission channel between the speech source and the receiver, wherein said predetermined function includes a first part having first parameters which models said source and a second part having second parameters which models said channel, wherein said processing step derives samples of at least said first parameters, and wherein said analyzing step determines values of said first parameters that are representative of said speech generated by said speech source before it was distorted by said transmission channel (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 49, Scholz et al., teach the method according to claim 48, wherein said function is in terms of a set of raw speech signal values, representative of speech generated by said source before being distorted by said transmission channel, further comprising a second processing step for processing the received set of signal values with initial estimates of said first and second parameters, to generate an estimate of the raw speech signal values corresponding to the received set of signal values and wherein said applying step applies said estimated set of raw speech signal values to said function in addition to said set of received signal values (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 50, Scholz et al., teach the method according to claim 49, wherein said second processing step uses a simulation smoother (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 51, Scholz et al., teach the method according to claim 49, wherein said second processing step uses a Kalman filter (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 52, Scholz et al., teach the method according to claim 48, wherein said second part is a moving average model and said second parameters comprise moving average model coefficients (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 53, Scholz et al., teach the method according to claim 37, further comprising the step of evaluating said probability density function for the set of received signal values using one or more of said drawn samples of parameter values for different numbers of parameter values, to determine respective probabilities that the predetermined signal model has those parameter values and wherein said processing step processes at least some of said drawn samples of parameter values and said evaluated probabilities to determine said values of said parameters that are representative of the received audio signal (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 54, Scholz et al., teach the method according to claim 37, wherein said analyzing step analyzes at least some of said derived samples of parameter values to determine a measure of the variance of at least some of said samples of parameter values, further comprising the steps of determining an indication of the quality of the received audio signal and wherein said encoding step encodes said determined

parameter values in dependence upon the determined quality indication (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 55, Scholz et al., teach the method according to claim 54, wherein said encoding step encodes said parameter values using a first encoding technique of said quality indication is above a predetermined values and encodes said parameter values using a second encoding technique if said quality indication is below said value (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 56, Scholz et al., teach the method according to claim 55, wherein said first encoding technique is operable to minimize the data to be transmitted and wherein said second encoding technique is operable to minimize information lost in the encoding (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 57, Scholz et al., teach an audio transmission method comprising the steps of: receiving an audio signal at a transmission unit, encoding the audio signal using a method according to claim 37 to generate encoded parameter values representative of the audio signal and, transmitting the encoded parameter values, receiving the transmitted encoded parameter values at a receiver unit, decoding the received encoded parameter values to generate and output signal in dependence thereon (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 58, Scholz et al., teach the method according to claim 37, wherein said processing step at said receiving unit comprises speech synthesis means for generating a synthesized speech signal in dependence upon the received parameter values (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 59, Scholz et al., teach the method according to claim 37, wherein said processing step at said receiving unit uses a speech recognition system to compare the received parameter values with stored reference models and to generate a recognition result (Col.9, line 67 – Col.10, line 7, Fig.2, Col.1, lines 16-22).

As per claim 60, Scholz et al., teach the method according to claim 57, further comprising the step of transmitting said quality indication to said receiving unit and, at said receiving unit, the step of receiving said quality indication and decoding said encoded parameters in dependence upon the received quality indication (Col.18, line 62 – Col.19, line 2).

As per claim 61, Scholz et al., teach the method according to claim 60, comprising the step at the receiving unit, decoding said encoded parameter values in accordance with a first decoding technique if said quality indication has a value above a predetermined threshold value and decoding said encoded parameter value in accordance with a second decoding technique if said quality indication is below said predetermined value (Col.18, line 62 – Col.19, line 2).

As per claim 64, Scholz et al., teach the method according to claim 61, wherein said encoding step encodes said audio signal in accordance with a first encoding technique if said quality measure is above a predetermined threshold and encodes said audio signal in accordance with a second encoding technique if said quality measure is below a predetermined threshold (Col.18, line 62, Col.12, line 2).

As per claim 67, Scholz et al., teach a computer readable medium storing computer executable process steps to cause a programmable computer apparatus to

perform the method of claim 37 (Col.11, line 67 – Col.12, line 7, Col.9, line 67, Col.10, line 7, Fig. 2, Col.1, lines 16-22, Col.10, lines 1-18, Col.12, lines 34-41, Col.10, lines 31-39, Col.12, lines 42-48, Col.11, lines 13-49, Col.18, line 62 – Col.19, line 2).

As per claim 68, Scholz et al., teach a processor implementable process steps for causing a programmable computing device to perform the method according to claim 37 (Col.11, line 67 – Col.12, line 7, Col.9, line 67, Col.10, line 7, Fig. 2, Col.1, lines 16-22, Col.10, lines 1-18, Col.12, lines 34-41, Col.10, lines 31-39, Col.12, lines 42-48, Col.11, lines 13-49, Col.18, line 62 – Col.19, line 2).

Conclusion

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Malvar (6,253,165) teaches a system and method for modeling probability distribution functions of transform coefficients of an encoded signal.

Malvar (6,256,608) teaches a system and method for entropy encoding quantized transform coefficients of a audio signal.

Haimi-Cohen (6,374,221) teaches automatic retraining of a speech recognizer while using reliable transcripts.

Vary et al., (6,546,515) teach a method of encoding a speech or an image signal.

Weerackody et al., (6,760,699) teach a soft feature decoding in a distributed automatic speech recognition system for use over wireless channels.

Davis (5,633,981) teaches a method and apparatus for adjusting dynamic range and gain in an encoder/decoder for multidimensional sound fields.

Handel (6,324,502) teaches noisy speech autoregression parameter enhancement method and apparatus.

Bartkowiak et al., (5,507,037) teaches an apparatus and method for discriminating signal noise from saturated signals and from high amplitude signals.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Vijay B. Chawan whose telephone number is (571) 272-7601. The examiner can normally be reached on Monday Through Friday 6:30-3:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571) 272-7602. The fax phone number for the organization where this application or proceeding is assigned is (571) 273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Vijay B. Chawan
Primary Examiner
Art Unit 2654

vbc
8/7/05

VIJAY CHAWAN
PRIMARY EXAMINER